CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 15, LINES 30-34

A1

Fig. 13 is the first basic IWDRC transducer in Fig. 12, with an adaptive compression threshold and providing the specified maximum low-level linear gain correction and intermediate-level power-law compression;

CLEAN COPY OF SPECIFICATION PARAGRAPH EXTENDING FROM PAGE 26, LINE 27 TO PAGE 27, LINE 3

A family of merging transfer functions, in accordance with the present invention, is obtained from $f(uG_i, u_i, p)$, wherein the instantaneous input amplitude, u, is amplified by G_i, and by requiring the following relationship between G_i and u_i:

AR

$$u_i = u_o G_i^{\frac{-p}{1-p}}$$

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 27, LINES 7-25

An alternative algorithmic implementation of the family of merging transducer functions is also provided, which is preferred when the basic transducer is realized with an optimized analog or digital module. The module maintains a fixed transducer function and uses pre- and post-amplification G_a and G_b that depend upon G_i:

$$G_a = G_i^{\frac{1}{1-p}}, \quad G_a G_b = G_i$$

Empirical constraints have been discovered for the smoothness of the transition from linear response to compressive response. Cochlear response functions for simple tone signals are well represented with the choice of the smoothness parameter, n, being set to 2. Pilot psychophysical study of speech intelligibility for normal-hearing listeners with the amplifier described in Fig. 1 have demonstrated significantly better performance (80% vs. 60%) when a smooth transition is provided between the linear region and compressive region (n=2), as opposed to a sharp transition (a sharp transition being a transition having a discontinuous derivative) wherein "n" is large ($n \to \infty$).

AB

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 29, LINES 6-28

Fig. 5 shows the required nonlinear gain corrections, for both the mildly impaired cochlea and the moderately impaired cochlea of Fig. 4. The gain correction required for the mildly impaired cochlea is represented by curve 208 while the gain correction required for the moderately impaired cochlea is represented by curve 210. These curves are derived from Fig. 4 by noting the horizontal distance in dB between the responses of the healthy and the impaired cochleae at the signal levels in dB shown. For example, at 20 dB SPL in Fig. 4, curve 200, representing the response of a healthy cochlea, shows a displacement of about 2.5 nanometers. A gain of slightly less than 40 dB is required to provide the same displacement for the moderately impaired cochlea, while a gain of only 20 dB is required for the mildly impaired cochlea. At 40 dB SPL, a gain of slightly less than 30 dB is required for the moderately impaired cochlea, while a gain of 20 dB still suffices for the mildly impaired cochlea. At about 60 dB SPL, the gain required for both the mildly and the moderately impaired cochlea is about 20 dB. As can be seen at greater SPLs, the required gain is essentially the same for both the mildly and moderately impaired cochlea, and this gain diminishes as SPL increases, approaching 0 dB for levels above approximately 100 dB SPL.

A4

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 33, LINES 13-22

A5

It should be noted that while the effects of shifting compression thresholds shown in Figs. 7-9 are general, the detailed calibration of the abscissa is dependent upon the division of the audio spectrum into separate bands in the amplifier design. In the six-channel amplifier that was investigated, most of the signal energy was divided among the lowest four octave bands. Therefore, to achieve similar results, a single channel wideband adaptive nonlinear amplifier would require over 6 dB greater shifts in compression threshold relative to the input signal.

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 35, LINES 3-15

This relationship is shown in Fig. 11, both graphically and analytically. The slightly modified gain specification G'(u) shown in Fig. 10 by the dotted lines is obtained by multiplying G(U) by the reciprocal of the describing-function factor, D(p):

$$D(p)=\left(\frac{2}{\sqrt{\Pi}}\right)\frac{\Gamma(1+0.5p)}{\Gamma(1.5+0.5p)2^{0.5(1-p)}};$$

A φ

wherein $\Gamma(\ )$ is the gamma function

This modification is further defined as equivalent to shifting the nonlinear thresholds to slightly higher values, as follows:

$$U_1'=U_1D(1/3)^{-3/2},\ U_2'=U_2D(1/3)^{-3/2},\ U_3'=U_3D(1/4)^{-4/3}$$

CLEAN COPY OF SPECIFICATION PARAGRAPH EXTENDING FROM PAGE 40, LINE 22 TO PAGE 41, LINE 7

The data set provided to the next lower channel 296 (2-4 kHz being the preferred range) is obtained by passing $S_i[n]$ through filter 292 which eliminates the frequencies in the highest range (the frequency range of channel 294) by means of lowpass digital filtering, and then downsamples the filtered data set by eliminating every second sample. Downsampled signal 298 is processed by the nonlinear transducer 108 of channel 296, with pre- and post-bandpass filtering (BPFs 104 and 112). Then, the signal leaving channel 296 is upsampled to the original sampling rate by lowpass and interpolation filter 302 and added to the output of the other channels. The upsampling is accomplished by inserting a zero amplitude sample after every second sample in the output of the second filter of the channel, and then interpolating the inserted sample amplitudes using lowpass filtering with amplitude scaling of 2. This scheme is repeated successively for each lower octave channel. The outputs of each channel are summed through the interpolation filters 302 and adders 304 to generate the resultant amplified sound signal Sum[n] provided to DAC 306.

A11

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 41, LINES 8-24

A8

In addition to the savings in data processing, the multirate design allows use of identical bandpass and lowpass filters in all channels. Many conventional techniques are available for filter design. A preferred design uses 21 tap FIR bandpass filters (with cutoff frequencies $\pi/4$ and $\pi/2$, and 22 tap lowpass filters (with cutoff frequency 0.30π), each synthesized as a windowed Butterworth IIR filter. The equalization stages 298 shown in Fig. 21 for the upper channels (all but the lowest frequency channel 300) are delays added to provide equal average group delay for the signals processed in each channel. Thus, a broadband audio signal with a well defined temporal epoch will be compactly reconstructed in a similar, but delayed, time window in the multichannel output Sum[n]. In an alternative design with a single channel broadband compressive audio amplifier, a linear phase filter design is preferred.

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 44, LINES 1-9

A9

Therefore, it is desirable that the compression threshold not be reduced until the signal's average sound level drops by a triggering amount, which identifies when a need exists to drop the compression threshold to adapt to a quieter environment. However, it is also desirable that the compression threshold quickly track increases in the sound signal's average sound level to minimize overamplification of background noise. The flowchart describes how these goals are accomplished.

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 44, LINES 21-32

A10

At step 1004, controller determines the peak (maximum magnitude) X_i of signal block $S_i[n]$. X_i will be the sample of $S_i[n]$ having the largest amplitude. Next, at step 1006, the controller will sort X_i with respect to the currently stored value of X. Essentially, the controller will determine (1) whether the peak is increasing from the stored peak (is $X_i > X$?), (2) whether the peak is decreasing an insignificant amount (is $\rho X < X_i \leq X$?, and (3) whether the peak is decreasing a significant amount, that is, decreasing by a triggering amount (is $X_i \leq \rho X$?). The parameter ρ is used to control the triggering amount. Preferably, $0<\rho<1$, and more preferably ρ is set equal to ½.

CLEAN COPY OF SPECIFICATION PARAGRAPH ON PAGE 50, LINES 7-23

A11

It is preferred to choose a common compression ratio (1/p) for all of the channels, so that the quiescent transducer responses for the different channels merge at high signal levels, while differing at low levels only in the compression threshold determined by G_1. The smallest compression ratio should be chosen from among the values 2, 3 and 4, to provide the range compression needed for signal frequencies of 0.5, 1.0 and 2.0 kHz. These frequencies are found to be most important for speech communication. Greater hearing losses at other frequencies should be corrected only to the extent possible with the chosen compression ratio. Compensation should be included for the loss of normal free-field acoustic amplification by the outer ear caused by use of standard earmolds or insert earphones. A preferred compensation provides a constant 14 dB gain emphasis for the 2-4 kHz channel relative to the other channels.